TITLE OF THE INVENTION

STRUCTURE OF EQUALIZER INTERACTING WITH TCM

BACKGROUND OF THE INVENTION

The present invention is to enhance performance of the equalizer of the ATSC 8-VSB receiver which is the U.S. terrestrial digital television standard, and relates to a method to enhance the performance of the equalizer by inputting an output of TCM (Trellis Coded modulation) instead of decision result, into the feedback filter of the general DFE (Decision Feedback Equalizer).

The conventional DFE (Decision Feedback Equalizer) consists of:

1. an FF (Feed-forward filter) unit to eliminate pre-echo influence,
2. an FB (Feedback filter) unit to eliminate post-echo influence,
3. a decision unit (8-level slicer) to determine an equalizer output of adding the FF unit and the FB unit as 8-level,
4. an Error Calculator unit to calculate error based on the decision result and the equalizer output, and
5. a coefficient updating unit to update filter coefficients according to the error and a preset step size.

Conventional Operation

The operation of the conventional DFE is as the following. (FIG. 1)

1. The channel equalization by the interacting between the FF (Feed-forward) unit having M taps and the FB (Feedback) unit having L taps.
2. Decision of convolution results y_n of the FF and FB units for 8-Level Slicer.
3. Input of decision results $\hat{y}_n$ into the FB unit and the error calculator.
4. Calculation of an error e_n according to LMS or Blind algorithm from $\hat{y}_n$ and y_n by the error calculator.
5. Input of the calculated error e_n to update coefficients of the equalizer.

The operation of the conventional DFE is described in the following. (FIG. 1)

Problems of the Prior Art

In the conventional DFE, the decision error is input to the feedback filter so that error propagation phenomenon may occur to degrade the equalization performance of the equalizer.

SUMMARY OF THE INVENTION

Object of The Invention

An object of the present invention is to enhance the performance of the equalizer by reducing the error propagation phenomenon which occurred in the conventional DFE (Decision Feedback Equalizer), of the feedback filter. More specifically, the equalization performance can be enhanced through solving the TCM delay problems which occur in

interacting with the TCM.

Construction of the Invention

The DFE (Decision Feedback Equalizer) of the present invention consists of:

1. a first equalizer to equalize and Trellis-decode an input signal of the equalizer,
2. a buffer to store an input from a second equalizer in consideration of delays of the first equalizer and the TCM, and
3. a second equalizer to equalize the input signal from the buffer by using the decoding result of the first equalizer (see FIG. 2).

The first equalizer may have any one of the three structures below.

1. General DFE structure [FIG. 1]
2. Structure in which single output of the TCM of the general DFE is input to the FB unit [FIG. 3]
3. Structure in which multi-output of the TCM of the general DFE is input to the FB unit [FIG. 4]

The second equalizer may have any one of the two structures below.

1. General DFE structure [FIG. 1]
2. DFE structure in which error is updated using Trellis-decoding results [FIG. 5]

A filter tap length of the first and second equalizers can be set appropriately considering the size of the hardware.

Operation of The Invention

Operations of the present invention are as follow. (FIG. 2)

In the first equalizer,

1. channel equalization is performed through the interacting between the FF (Feed-forward) unit and the FB (Feedback) unit.
2. convolution results of the FF unit and the FB unit is decided with the 8-level slicer.
3. the decision result is input to FB unit and the error calculator.
4. the error calculator calculates errors according to the LMS or the Blind algorithm.
5. the calculated error is input into the FF and FB units to update the coefficients of the equalizer.

In the TCM, Trellis decoding is performed from an output of the first equalizer.

FIG. 6 shows Data Recycling Performance of Brazil A channel.

The buffer, which has a length of adding the first equalizer delay and the TCM delay, stores the inputs of the equalizer and outputs to the second equalizer in time-sync with the outputs of the first equalizer.

In the second equalizer,

1. the channel equalization is performed through the interacting between the FF (Feed-forward) unit and the FB (feedback) unit.
2. convolution results of the FF and FB units are decided with the 8-level slicer.
3. a TCM result is input to the FB unit and the error calculator.

4. the error calculator calculates errors according to the LMS or the Blind algorithm.
5. the calculated error is input to the FF and FB units to update coefficients of the equalizer.

Effect of The Invention

By using the present invention, errors contained in a signal input to the feedback unit are greatly reduced through the TCM so that the performance degradation due to the error propagation is prevented. Also, by addressing the TCM delay problems occurring in interacting with the TCM, the performance of the equalizer can be enhanced.

Further, in the second equalizer, errors can be accurately calculated by using output of the TCM which is superior to the convolution results of the FF and FB units in view of SER. As a result, the performance of the equalizer is enhanced.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. 1 shows a general DFE structure,

FIG. 2 shows an interaction structure of the TCM and the equalizer,

FIG. 3 shows an equalizer structure interacting with the single output TCM,

FIG. 4 shows an equalizer structure interacting with the multi output TCM, and

FIG. 5 shows an equalizer structure updating errors by using the Trellis-decoding result.

Drawing 1 shows VSB data frame structure,

Drawing 2 shows field synchronization signal format,

Drawing 3 shows block diagram of VSB modulator,

Drawing 4 shows decision feedback equalizer,

Drawing 5 shows data input method in data recycling,

Drawing 6 shows data recycling performance (Brazil channel A),

Drawing 7 shows data recycling performance (Brazil channel B),

Drawing 8 shows data recycling performance (Brazil channel C), and

Drawing 9 shows data recycling performance (Brazil channel D).

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS

A Design of 8-VSB Equalizer

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ABSTRACT

This paper relates to the DFE (Decision Feedback Equalizer) corresponding to the equalizer among receivers of 8-VSB (Vestigial Sideband) system which is the U.S. terrestrial digital system, and illustrates the summary of the DFE structure and performance analysis results of the Blind algorithm. We suggest an optimized structure of the equalizer to improve its performance in an environment such as Brazil where indoor reception is inferior, and performance analysis results obtained through conducting simulations after

implementing an interaction method of the equalizer and TCM and a filter coefficient initialization algorithm.

1. Summary

In early 1996, the Korean government announced the terrestrial broadcasting digitalization plan, selected the U.S. system in November of 1997, and confirmed the standard in August of 1998. Through a test broadcasting in 2000, a full-scale broadcasting is intended to launch in 2001. We tested the ATSC (Advanced Television System Committee) system selected as the domestic terrestrial digital broadcasting standard, with the receiver of early model. As a result, the broadcast reception in a city is poorer than the current analogue system and is frequently infeasible in a moment when receiving through an indoor antenna. Thus, viewer experiences inconvenience and it made social issues.

In U.S.A, the Sinclair Broadcast Group has examined the reception state of the ATSC system terrestrial broadcast and pointed out that the antenna in a city needs minute adjustment. Also, there arise problems in the reception if a vehicle passes or a tall building is located in front of the antenna.

To overcome those problems, various standards and plans for improving the broadcast reception have been presented to enhance the reception performance.

Chapter 2 of this paper illustrates the summary of the VSB system which is the U.S. digital TV system. Chapter 3 illustrates the DFE structure of the equalizer contained in the

VSB receiver and the blind algorithm. Chapter 4 illustrates the structure of our suggested equalizer and an equalization method to improve the reception performance in the Brazil channels which is a poor environment where most of the 8-VSB receivers fail to receive the broadcast. Chapter 5 illustrates the structure of the equalizer interacting with the TCM to reduce the error propagation influence which may occur in the DFE structure. Chapter 6 illustrates an initialization algorithm to improve a convergence speed of the equalizer, and chapter 7 is the conclusion.

2. Summary of VSB system for digital TV

A VSB signal is transmitted in a frame unit as shown in FIG. 1[1]. Each frame consists of two fields, and each field consists of a single field synchronization segment and 312 data segments. Each of the data segments consists of 832 symbols. First 4 symbols are segment synchronization symbols having values of 5, -5, -5, and 5. The other 828 symbols are data symbols having levels of ∓ 1, ∓ 3, ∓ 5, and ∓ 7. The field synchronization segment of FIG. 2 also consists of 832 symbols of which 4 segment data symbols are followed by 704 training sequences. The training sequences consist of a binary PN (Pseudo Noise) sequence which is 63 in length and 3 binary PN sequences which is 63 in length. The second 63-PN sequence of the 3 63-PN sequences outputs an inversed signal in every field. In addition, to facilitate the demodulation, a pilot is generated by adding 1.25 to the 8 signal levels. The segment synchronization signal and the field synchronization do not go through

encoding and interleaving, and are inserted to data stream prior to the pilot signal insertion and modulation.

FIG. 3 is a block diagram of the VSB modulation process. A data randomizer performs XOR operation with 16-bit PRBS (Pseudo Random Binary Sequence) in respect to the entire data except for the segment synchronization signal and the field synchronization signal. The PRBS consists of 16-bit shift register and has 9 feedback taps. 8 of the outputs from the shift register are outputs of the data randomizer. The generator polynomial is $g(x) = x^{16} + x^{13} + x^{12} + x^{11} + x^{7} + x^{6} + x^{3} + x + 1$. A RS (Reed Solomon) code uses (207, 187) code capable of correcting 10-bit error, and the primitive polynomial is $x^8 + x^4 + x^3 + x^2 + x + 1$. The RS-encoded data is input to a 2/3 rate Trellis interleaver through a (52, 208) convolution bite interleaver so as to prevent burst errors. The 8-level data stream finally generated has 10.76MHz symbol rate, and is modulated and transmitted after passing through the VSB modulation filter which has 6MGz band of which the roll-off factor is 11.5%. The receiver passes through the reverse process of the modulator. A channel equalizer is added to compensate signal distortion due to the channel.

3. Structure of general channel equalizer and equalization method thereof

3.1 Structure of channel equalizer

The channel equalizer can be implemented as a transversal structure or a lattice

structure according to the filter structure. The transversal structure provides easy implementation of hardware compared with the lattice structure, and the lattice structure is superior in view of error convergence.

The filter of the transversal structure can be divided into a general linear structure and a decision feedback structure of FIG. 4. The DFE consists of a feed-forward unit which is a FIR (finite impulse response) filter and a feedback unit which is an IIR (infinite impulse response) filter. The feed-forward unit eliminates pre-ghost influence among input signals received through multipaths, and the feedback unit eliminates post-ghost influence. Through the interaction of the feed-forward unit and the feedback unit, the DFE can effectively compensate the channel distortion with a less taps than the linear equalizer consisting of the feed-forward unit alone. An output $\hat{I}_K$ of the equalizer is expressed as follows.

$$\hat{I}_K = \sum_{n=-K}^{0} f_n y_{k-n} + \sum_{n=1}^{L} b_n I_{k-n} \qquad (1)$$

Here, y_{k-K} denotes the input signal to the equalizer, and I_{k-n} denotes the output passed through the 8-level slicer. f_n, b_n respectively denote the tap coefficients of the feed-forward unit and the feedback unit.

3.2 Adaptive equalization method

In the channel coefficient adaptation process, it is necessary to obtain errors between a desired signal and the filtered signal. However, the accurate error value is not obtained since the receiver cannot get the transmitted signal value. Thus, the performance of the

application process is deteriorated. These problems can be overcome by applying coefficients using the training sequence, which is a special bit sequence known to both of the sender and the receiver in a preset cycle.

However, the above solution cannot cope with the fast-changing channels. Also, there is a problem in efficiency since unnecessary data have to be added to the transmitted stream. To settle these problems, a blind adaptive equalization algorithm has been proposed [3][4], which estimates the transmitted signal value only with data without the training sequence.

In case of the ATSC 8-VSB, 704 training sequences are periodically transmitted at every 260,000 data symbol. Thus, the equalizer of the receiver operates in LMS (Least Mean Square) while the training sequences are input, and operates in the blind adaptive equalization algorithm of adapting the equalizer with the training sequences, while the data are input.

3.3 Blind adaptive equalization algorithm

3.3.1 Decision Directed algorithm

The Decision Directed algorithm substitutes the reference signal to substitute the training sequence by a level value of the transmitted signal which is the closest to the output [2]. That is, in the error equation of the Decision Directed algorithm, the decision result $I(k)$ becomes the transmitted signal closest to the equalizer output $\hat{I}(k)$.

$$e_{DD}(k) = \hat{I}(k) - I(k) \qquad (2)$$

The Decision Directed algorithm has advantages in that MSE (Mean Square Error) after the convergence is lower than other blind algorithms.

3.3.2 G-pseudo algorithm

At early stages, since the decision error $e_{DD}(k)$ is greater than the blind error $e_{Blind}(k)$ before the equalizer converges, coefficients are updated using the blind error. After the convergence, the coefficients are updated using the decision error [3].

$$e_{G-pseudo}(k) = k1 \cdot e_{DD}(k) + k2 \cdot |e_{DD}(k)| \cdot e_{Blind}(k) \quad (3)$$

In the above equation obtaining the G-pseudo error, the convergence speed and the residual errors can be adjusted by substituting $k1$, $k2$ by proper constant values. Generally, if $k1$ is relatively greater than $k2$, the residual error is less, and if the $k2$ is relatively greater, the convergence speed is high.

3.3.3 Stop-and-Go algorithm

The Stop-and-Go algorithm allows reliability to errors for the coefficient update, and updates the coefficients only if signals of the decision error and the blind error are the same [4].

$$e_{SAG}(k) = f(k) \cdot e_{DD}(k) \quad (4)$$

$$f(k) = \begin{cases} 1, & if \ \text{sgn}\ (e_{DD}(k)) = \text{sgn}\ (e_{Blind}(k)) \\ 0, & if \ \text{sgn}\ (e_{DD}(k)) \neq \text{sgn}\ (e_{Blind}(k)) \end{cases}$$

The Stop-and-Go algorithm guarantees the convergence as in using the blind error, but has shortcomings in that the convergence speed is lower due to the low frequency of updating coefficients. Meanwhile, in updating the coefficients, since the Stop-and-Go

algorithm uses the decision error, the MSE after the convergence becomes less to almost the same level of the Decision Directed algorithm.

In comparing the G-pseudo algorithm with the Stop-and-Go algorithm, the latter has the low convergence speed but stably operates in such a poor environment with the simple method. Also, when converged, the MSE level is lower than the G-pseudo algorithm [2].

4. Structure of the suggested equalizer

4.1 Overlapped DFE structure

The DFE structure having the equalizer main tap located at f_0 has advantages in that hardware complexity is low, and, in poor channel environment, can implement a stable system capable of eliminating the pre-ghost and the post-ghost with the feed-forward unit alone, by extending the number of taps of the post-ghost side based on the main tap of the feed-forward unit. That is, the feed-forward unit, i.e., the FIR filter, is relatively stable comparing with the feedback unit, i.e., the IIR filter. Hence, it is preferred that the feed-forward unit converges to some degree at an early state or in the poor environment so that the IIR filter, i.e., the feedback unit, can stably converge.

4.2 Performances of the general DFE and the Overlapped DFE

To compare the performances of the general DFE structure and the Overlapped DFE structure, we conducted the computer simulation with respect to the Brazil channels,

which is the indoor reception environment having serious channel distortion due to multipaths.

In the simulation, the lengths of the equalizer taps in the feed-forward unit and the feedback unit are 512 taps. In simulating the Overlapped DFE structure, half of the feed-forward taps, i.e., 256 taps are overlapped by setting the main tap in the middle of the feed-forward unit. As for the Blind algorithm, the Stop-and-Go algorithm is used.

Table 1. Comparison of performances of DFE and Overlapped DFE

	output SNR (dB)		Convergence (ms)	
Ch.	DFE	over. DFE	DFE	over. DFE
A	13.3	13.5	0.0377	0.0143
B	11.8	12.2	0.1392	0.0585
C0	13.5	14.2	0.3266	0.2055
C1	13.6	14.7	0.2716	0.1223
D1	12.0	13.0	0.2551	0.1613
D2	7.38	14.3	x	0.1144
E	9.4	10.3	x	0.4281

(C0, D1 are channels setting first multipaths of channel 'C', 'D' as the main. C1, D2 are channels setting the larges multipaths as the main. E1 is a channel changing the delay of the path2 in E0 channel from 1.0us to 1.1us [Attachment 1].)

According to the simulation results, the output SNR (Signal to Noise Ratio) and the convergence speed of the equalizer of the general DFE structure are lower than those of the equalizer of the Overlapped DFE structure.

4.3 Filter tap length and step size

The filter tap length of the equalizer is a significant factor determining the performance of the equalizer in relation to the range of the ghost that the equalizer can eliminate. Our suggested equalizer consists of the feed-forward consisting of 512 taps and the feedback consisting of 512 taps, and has the structure using the overlapped feed-forward consisting of 192 taps. That is, based on the main tap, there are 320 taps on the pre-ghost side and 512 taps on the post-ghost taps, and the ghost influences –29.7usec ~ 47.5usec can be compensated.

$$0 \langle \mu = \frac{1}{tap-input\ \ power} \cdot \frac{1}{\alpha} \langle \frac{2}{tap-input\ \ power} \qquad (5)$$

The step-size of the equalizer is a variable determining the convergence speed of the equalizer and the error level after the convergence, and defined as the above equation. According to the constant α, a proper step size is determined and used. Generally, if the step-size is greater, the convergence speed is faster and the error rate after the convergence increases. If the step size is smaller, the convergence speed is slower and the error rate after the convergence decreases [2].

To obtain the step-size having the optimal performance, we analyzed the equalizer performance according the step-size ratio of the feed-forward and the feedback filters. Table 2 shows values measuring the output SNR when the input SNR is 15dB in the Brazil channels, and indicates that the greater the value, the better the performance.

According to the simulation results, the performance is superior when the feed-forward step-size is twice as large as the feedback step-size. Thus, the step-size of the

suggested equalizer uses multiples of two as the feed-forward unit 3.8e-6(2-18) and the feedback unit 1.9e-6(2-19) to facilitate the implementation in designing the hardware.

Further, in order to improve the convergence speed of the equalizer, we designed the variable step-size structure capable of reaching a high convergence level by enlarging the step-size at the early state and lessening the step-size after converging beyond a preset level. That is, at the early state when the output SNR is less than 14dB, we equalize fast by using the step-sizes four times as large as 3.8e-6, 1.9e-6, and when the output SNR is greater than 14dB, we equalize using 3.8e-6 and 1.9e-6.

5. Structure of equalizer interacting with TCM

The DFE structured equalizer has less taps and faster convergence speed than the Linear structure. But, accuracy of estimation signals input to the feedback filter determines the performance of the equalizer. In other words, if inaccurate estimation signal are input to the feedback filter due to noise, the error propagation phenomenon may occur so that the channel distortion due to the multipaths is not compensated [5]. Since the error propagation phenomenon gets worse as the channel environment is inferior, the accurate estimation on signals is required. Especially, in case of the equalizer having quite long taps like the suggested equalizer, decision error components remain at the filter for a long time so as to raise probability to adversely affect the performance of the equalizer. Thus, we suggest the structure of the equalizer interacting with the multi output TCM to enhance the performance

of the equalizer by interacting the equalizer with the TCM process included in the 8-VSB standard.

Table 2. Performance of equalizer according to feed-forward/feedback step-size ratio

Feed-back	Feed-forward	A	B	C0	C1	D1	D2	E
$\alpha = 50$	$\alpha = 10$	13.7862	12.2115	13.7838	13.7981	13.7228	13.8971	9.5935
	$\alpha = 20$	14.0339	12.4509	14.0705	14.0411	13.9833	14.1533	9.7463
	$\alpha = 30$	14.1374	12.5427	14.1352	14.1183	14.0390	14.2567	9.7750
	$\alpha = 50$	14.1823	12.5278	14.0761	14.0934	14.0317	14.2645	9.7524
	$\alpha = 100$	14.2281	12.4921	13.7416	13.9083	13.8114	13.9094	9.5853
$\alpha = 100$	$\alpha = 10$	13.8312	12.2761	13.9005	13.8175	13.7688	13.9508	9.6367
	$\alpha = 20$	14.0971	12.5148	14.1175	14.1412	14.0839	14.2005	9.7985
	$\alpha = 30$	14.1977	12.5795	14.1972	14.2136	14.1418	14.2945	9.8341
	$\alpha = 50$	14.2444	12.6294	14.1901	14.2391	14.1606	14.3494	9.8395
	$\alpha = 100$	14.2996	12.6061	13.9222	x	13.9222	13.9018	9.6550

5.1 Trace back delay of TCM decoder

The TCM used in the 8-VSB carries out TCM encoding to output data and gives interleaving effects in 12 symbol units by using 12 identical Trellis encoders having 2/3 rate and precoders. The TCM decoder of the receiver decodes using the 8-level Viterbi decoding algorithm [6].

Generally, because the performance of the TCM decoder is in proportion to decoding depth calculating survival path, the reliability of signals is elevated as the decoding depth lengthens [7]. However, as the decoding depth lengthens, the trace back delay of the

TCM outputs to the feedback filter of the equalizer also lengthens. That is, in the structure in which the final output of the TCM decoder having a preset decoding depth (N) is output to the feedback filter of the equalizer, the feedback filter tap corresponding to the trace back delay symbol length is input with the decision results of the equalizer, and only the feedback filter after the trace back delay symbol length is input with the TCM output. Accordingly, in the channel environment having strong ghost within the trace back delay symbol length, it is hard to expect the performance improvement due to the interaction of the equalizer and the TCM.

The suggested equalizer improves the performance of the equalizer in the reception environment such as Brazil having the strong ghost within the trace back delay symbol, by interacting the equalizer with the Multi output TCM and thus minimize the influence of the trace back delay.

5.2 Multi output TCM interacting equalizer

In case of the 8-VSB standard, because input data is interleaved in 12-symbol unit, the TCM decoder used in the receiver also outputs the estimation signals having relatively high reliability in 12-symbol unit.

The Multi output TCM is a TCM decoder using the output characteristics of the VSB system TCM decoder and outputting transmitted signal estimation results from each decoding state of the TCM. That is, while the conventional TCM decoder only outputs the estimation results after a preset decoding depth (N), the Multi output TCM outputs the

estimation results in 12-symbol unit from each state in which the decoding depth are respectively 0, 1, 2, ..., N. The method of interacting the Multi output TCM and the equalizer is to input the output of the state in which the decoding depth is 0 into the 1st tap of the feedback filter, and the output of the state in which the decoding depth is 1 into the 12th tap of the feedback filter, and input the outputs of the states into the feedback filters in 12-symbol unit up to outputs of a preset decoding depth. In interacting the equalizer with the Multi output TCM, the feedback filter is input with the signals having error rate lower than the decision results. Therefore, the error propagation can be prevented and the performance of the equalizer can be improved even in a poor channel environment.

5.3 Performance of equalizer interacting with Multi output TCM

We tested the performance of the equalizer structure interacting with the Multi output TCM to prevent the error propagation phenomenon of the feedback filter of the equalizer.

The test results in the Brazil channels show that, in case of the equalizer interacting with the Multi output TCM, TOV (Threshold of Visibility) is lower at most by 4.7dB in comparison with the equalizer not interacting with the TCM. Also, the Brazil E channel, i.e., the SFN (Single Frequency Network) channel model, which failed to receive the broadcast while not interacting with the TCM, received the broadcast by interacting with the Multi output TCM.

Table 3 shows the performance of the equalizer, which is equal to or superior to the

highest performance in the Brazil channels, of the best receiver among the currently released VSB receivers [8].

Table 3. Input SNR (dB) at TOV

Brazil Channel	Overlapped DFE Only	Overlapped DFE + TCM
A	16.4	15.9
B	18.6	16.0
C0	15.4	13.4
C1	15.0	13.3
D1	18.2	13.5
D2	15.4	13.0
E0	x	18.2
E1	15.1	13.5

(C0, D1 are channels setting first multipaths of channel 'C', 'D' as the main. C1, D2 are channels setting the larges multipaths as the main. E1 is a channel changing the delay of the path2 in E0 channel from 1.0us to 1.1us [Attachment 1].)

6. Equalizer initialization algorithm

The performance of the channel equalizer depends on the output SNR level after the convergence, and the convergence speed. Recently, a solution to improve the convergence speed of the equalizer has been presented, which uses training sequences included in transmitted data stream [9]. This paper proposes Data Recycling algorithm for the equalizer initialization algorithm.

6.1 Data Recycling algorithm

In the ATSC 8-VSB standard, training sequences are transmitted in every 260,000 symbols. As a result, the equalizer has to operate in a blind mode from after the training sequences are initially input until next training sequences are input. In the poor channel environment, even if the equalizer diverges or not in the initial blind mode, errors are cumulated to lower the convergence speed.

The data recycling is an initialization algorithm which improves the convergence speed by repeatedly using the training sequences at the early state of the equalizer and initializing tap coefficients. FIG. 5 illustrates the input method of data during the data recycling. Specifically, if the training sequences are initially input, the input training sequences are stored and used continuously for the equalization while the equalizer is not converted to the blind mode until reaching a preset level of the convergence. The usage of the Data Recycling algorithm should be limited to get SNRs over a preset level so that the Blind algorithm operates normally.

6.2 Performance of Data Recycling algorithm

Carrying out the data recycling several times provides the same effect as the equalizer provides in noise-free environment. In detail, as the equalizer is input with the identical values continuously, the equalizer carries out the equalization based on the determination that AWGN (Additive White Gaussian Noise) components included in the input are components from the multipaths. Accordingly, the data recycling more than a

preset number of times is unnecessary. Our proposed equalizer initializes the equalizer by using the data recycling 12 times.

In the Brazil channels, we compared the convergence speeds of the equalizer in the cases of using the initialization algorithm and using the Data Recycling algorithm.

According to the comparison results of FIGs. 6~9, in case of using the Data Recycling algorithm, the convergence speed increases at the early state with respect to all of the Brazil channels. As the variable step-size structure is used, the equalizer may diverge at the early stage in the environment where the channel distortion is severe as in the Brazil C, D channels. By using the data recycling, the equalizer can be operated stably while equalizing with a large step-size at the early stage.

7. Conclusion

The equalizer we proposed has the Overlapped DFE structure which uses 512 feed-forward filter taps, 512 feedback filter taps, and 320th tap of the feed-forward filter as the main tap, and its ghost elimination range is about –29.7usec ~ 47.5usec. As for the Blind algorithm, the Stop-and-Go algorithm is used.

In case that the signal distortion is severe due to the multipath such as the indoor reception for DTV, we proposed the interacting structure of the equalizer and the TCM to prevent the error propagation of the feedback filter. Especially, to minimize the influence by the trace back delay occurring while the equalizer interacts with the TCM, the

performance of the equalizer has been greatly improved by interacting with the Multi output TCM in the channel environment having the strong ghosts within the trace back delay symbol length.

To enhance the convergence speed of the equalizer, we used the variable step-size which increases the initial convergence speed. The filer taps of the equalizer were initialized using the Data Recycling algorithm.

Accordingly, we designed the equalizer having high performance in view of the output SNR level after the convergence and the convergence speed with respect to the ATSC 8-VSB standard.

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Attachment 1. Characteristics of Brazil channels

		Path 1	Path 2	Path 3	Path 4	Path 5	Path 6
A	Amplitude	1.0	0.2.45	0.1548	0.1790	0.2078	0.1509
	Relative [dB]	0	-13.2	-16.2	-14.9	-13.6	-16.4
	Delay [us]	0	0.15	2.22	3.05	5.86	5.93
B	Amplitude	1.0	0.2512	0.6310	0.4467	0.1778	0.0794
	Relative [dB]	0	-12.0	-4.0	-7.0	-15.0	-22.0
	Delay [us]	0	0.3	3.5	4.4	9.5	12.7
C0	Amplitude	0.7263	1.0	0.6457	0.9848	0.7456	0.8616
	Relative [dB]	-2.8	0	-3.8	-0.1	-2.5	-1.3
	Delay [us]	0	0.089	0.42	1.51	2.32	2.80
C1	Delay [us]	-0.089	0	0.33	1.417	2.233	2.71
D1	Amplitude	0.2045	0.1341	0.1548	0.1789	0.2077	0.1509
	Relative [dB]	-0.1	-3.8	-2.6	-1.3	0	-2.8
	Delay [us]	0	0.48	2.07	2.90	5.71	5.78
D2	Delay [us]	-5.71	-5.23	-3.64	-2.81	0	0.07
E0	Amplitude	1.0	1.0	1.0	OFF	OFF	OFF
	Relative [dB]	0	0	0	OFF	OFF	OFF
	Delay [us]	0	1	2	OFF	OFF	OFF
E1	Delay [us]	0	1.1	2	OFF	OFF	OFF

WHAT IS CLAIMED IS:

1. an equalizer structure interacting with TCM;

2. a first equalizer to equalize and Trellis-decode an input signal;

3. a buffer to consider the first equalizer and a Trellis-decoding delay;

4. a second equalizer using an input from the buffer and a Trellis-decoding output;

5. a DFE structured equalizer having a general structure of the first equalizer;

6. the DFE equalizer interacting with a single output TCM through the first equalizer structure;

7. the DFE equalizer interacting with a multi output TCM through the first equalizer structure;

8. the DFE structured equalizer having a general structure of the second equalizer;

9. the equalizer having a structure in which the second equalizer updates errors using the Trellis decoding result and the equalizer output.

DECLARATION

I, Ginny Kang, a Korean citizen of 916-13, Daechi-4-dong, Gangnam-gu, Seoul, Korea do hereby solemnly and sincerely declare as follows:

1. That I am well acquainted with the English and Korean languages.
2. That the following is a correct translation into English of the U.S Provisional Application No. 60/430,359 filed December 3, 2002, and I make the solemn declaration conscientiously believing the same to be true.

Seoul, December 1, 2003

Ginny Kang

Ginny Kang